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Assistant Commissioner for Patents
Washington, D.C. 20231

Sir:

Transmitted herewith for filing is the
Patent Application of:

      Inventor: Michael L. Honig

      For: REDUCED RANK ADAPTIVE FILTER

      Case Docket No.   12569US01

Enclosed are:

  X   4 Sheets of drawing (informal)

  X   10 Pages of specification, _7_ pages of claims, and _1_ page of Abstract.

  X   ____ Verified Statement (Declaration) Claiming Small Entity Status (37 C.F.R. 1.9(F) and 1.27(c)) Small Business Concern   (Unsigned)

  X   Declaration and Power of Attorney   (Unsigned)

Fees:
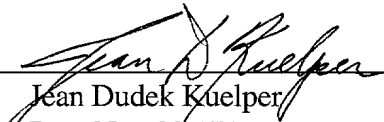
Claims As Filed	Extra Claims	Rate Per Extra Claim	
Basic Fee			$345.00
Total claims	25 - 20 = 5	x $9.00	$ 45.00
Indep. claims	4 - 3 = 1	x $39.00	$ 39.00
Multiple dependent claim presented ($230 if any present)			$
Total Filing Fee			$429.00

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Attorney Docket No.: 12569US01   PATENT

Applicant or Patentee: Michael L. Honig

Filed herewith for: REDUCED RANK ADAPTIVE FILTER

VERIFIED STATEMENT CLAIMING SMALL ENTITY STATUS
(37 CFR 1.27(d) - NONPROFIT ORGANIZATION)

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[   ]   the application identified above.
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*NOTE: Separate verified statements are required from each named person, concern or organization having rights to the invention averring to their status as small entities: (37 CFR 1.27).

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I hereby declare that all statements made herein of my own knowledge are true and that all statements made on information and belief are believed to be true; and further that these statements were made with the knowledge that willful false statements and the like so made are punishable by fine or imprisonment, or both, under section 1001 of Title 18 of the United States Code, and that such willful false statements may jeopardize the validity of the application, any patent issuing thereon, or any patent to which this verified statement is directed.

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REDUCED RANK ADAPTIVE FILTER

CROSS-REFERENCE TO RELATED APPLICATIONS

N/A

STATEMENT REGARDING FEDERALLY SPONSORED RESEARCH OR DEVELOPMENT

5   ### FIELD OF THE INVENTION

The present invention is directed to a method of adaptive digital filtering and more particularly to such a method utilizing a reduced rank technique.

10   ### BACKGROUND OF THE INVENTION

Adaptive digital filters are known for use in various signal processing applications including speech and radar processing, adaptive beamforming, echo cancellation and equalization. These filters estimate

15   the optimal filter coefficients from observed data, i.e. data representing the received signal to be processed.

Reduced rank filters estimate the filter coefficients with a relatively small amount of observed data. Various reduced rank techniques are known including Principal Components, Cross-Spectral

20   and Partial Despreading methods. The former two methods require an explicit estimate of the signal subspace via an eigen-decomposition of

- 2 -

the input covariance matrix which is extremely complex. Although the latter technique is much simpler, it does not achieve near full rank performance when the filter rank D is significantly less than the full rank N.

5   Other known reduced rank techniques include a multi-stage Wiener filter as proposed by Goldstein et al. in "A Multistage Representation Of The Wiener Filter Based On Orthogonal Projections" IEEE Trans. Inform. Theory, 44 (7), November, 1998 and an adaptive interference suppression algorithm as proposed by Honig et al. in

10   "Adaptive Reduced-Rank Residual Correlation Algorithms For DS-CDMA Interference Suppression" In Proc. 32 Asilomar Conf. Signals, Syst. Comput., November, 1998. These methods perform well and do not require an eigen-decomposition. The present invention is an improvement of these latter two methods.

15

BRIEF SUMMARY OF THE INVENTION

       In accordance with the present invention, the disadvantages of prior adaptive digital filters noted above have been overcome. The

20   reduced rank adaptive filter of the present invention projects a received signal onto a lower dimensional subspace where the reduced rank subspace is iteratively constructed by multiplying the last basis vector by the received sample covariance matrix. The reduced rank filter of the present invention achieves full rank performance with a projected signal

25   subspace having a significantly smaller dimension than the dimension of the received signal subspace. Moreover, for interference suppression applications, the optimum filter rank does not substantially increase with the dimension of the received signal subspace. As such, the method of the present invention enables faster tracking and convergence with

30   significantly less training samples than can be achieved with prior techniques.

More particularly, the method of the present invention filters successive, received signal samples to provide a desired signal wherein a group of N successive samples form a received sample vector of digital data having a N x 1 dimension. The method includes generating D + 1 basis vectors where D is less than N and represents the reduced rank dimension. Each successive basis vector is generated from a given or an estimated steering vector and successively greater powers of a covariance matrix for a sequence of received sample vectors of data. The first basis vector is formed from the steering vector. A reduced rank vector of digital data having a D x 1 dimension is generated from a matrix of D basis vectors and the received sample vector of data. D filter coefficients are generated from correlations between pairs of the basis vectors. The desired signal output from the filter is generated from the filter coefficients and the reduced rank vector of data.

The method of the present invention can be used in any adaptive filtering application including echo and noise cancellation, channel equalization, radar processing, adaptive antenna arrays and interference suppression. The method of the present invention is particularly advantageous for applications where long filter lengths are required and fast convergence and tracking are important. One such application is for space-time interference suppression in wireless Code-Division Multiple Access (CDMA) systems.

These and other advantages and novel features of the present invention, as well as details of an illustrated embodiment thereof, will be more fully understood from the following description and drawings.

BRIEF DESCRIPTION OF THE SEVERAL VIEWS OF THE DRAWING

Fig. 1 is a block diagram of the reduced rank adaptive filter of the present invention for a wireless CDMA communication system;

Fig. 2 is a flow chart illustrating the steps of the reduced rank adaptive digital filtering technique of Fig. 1;

Fig. 3 is a block diagram illustrating the generation of the received signal sample covariance matrix and steering vector;

Fig. 4 is a diagram illustrating the generation of the basis vectors; and

Fig. 5 is a flow chart illustrating the method of selecting the dimension of the reduced rank filter.

DETAILED DESCRIPTION OF THE INVENTION

The reduced rank adaptive digital filtering technique 10 of the present invention is illustrated in Fig. 1 for interference suppression in a wireless CDMA communication system having a receiver 12. Although illustrated for a wireless communication system, the filtering technique of the present invention can be used in any adaptive digital filter application. The receiver 12 picks up a received signal via an antenna 14 coupled to a radio frequency (RF) amplifier and mixer 16. The output of the amplifier/mixer 16 is coupled to an analog to digital converter 18 to provide a serial stream of baseband digital data samples representing the received signal. The output of the digital to analog converter 18 is coupled to a serial to parallel converter 20. The serial to parallel converter 20 takes N serial samples and outputs the N samples in parallel to provide a received sample vector $r(i)$ of digital data having a dimension of N x 1.

The received sample vector $r(i)$, which is referred to as an observation signal, is coupled to an adaptive digital filter 22, the output of which is an estimate of the desired signal $b(i)$, referred to as the approximate desired signal $\hat{b}(i)$. A generator 24 produces a matrix M of basis vectors with a N x D dimension where D is less than N and represents the reduced rank dimension. The generator 24 also produces

a vector of filter coefficients $\tilde{c}$ of dimension D x 1 from a sequence received sample vectors $r(i)$ for $i = 1, ..., B$ and from a sequence of the decision derived desired signals $b(i), i = 1, ..., B$ or a sequence of given desired signals or training symbols $b(i), i = 1, ..., B$. The given training

5   symbols $b(i)$ initially used by the generator 24 may be stored, for example, in a receiver memory (not shown).

The adaptive filter 22 generates a reduced rank or projected received signal vector $\tilde{r}(i)$ of dimension D x 1 by multiplying the Hermitian transpose of the matrix M by the received sample vector $r(i)$

10   as follows.

$$\tilde{r}(i) = M^{\dagger}r(i)$$

The adaptive filter 22 generates the approximate desired signal $\hat{b}(i)$ by multiplying the Hermitian transpose of the filter coefficient vector by the reduced rank vector as follows.

15
$$\hat{b}(i) = \tilde{c}^{\dagger}\tilde{r}(i)$$

The approximate desired signal $\hat{b}(i)$ is applied to a conventional slicer 23 that essentially rounds the approximate desired signal $\hat{b}(i)$ to a desired signal level $b(i)$ which may be used to generate the filter coefficients in a decision directed training mode.

20   The generator 24 generates D + 1 basis vectors where each successive basis vector is generated from a given or an estimated steering vector p and successively greater powers of a sample covariance matrix R for a sequence of B received sample vectors $r(i), i = 1, ..., B$. The first basis vector is formed from the steering vector p as described

25   in detail below with reference to Fig. 4. The generator 24 generates a D x 1 vector of filter coefficients $\tilde{c}$ from a matrix and vector of correlations between pairs of generated basis vectors.

The generator 24 and adaptive digital filter 22 operate in accordance with the method depicted in Fig. 2. At block 26, the generator 24 determines the sample covariance matrix R for the sequence of received sample vectors $r(i), i = 1, ..., B$ and the steering vector p from the sequence of received sample vectors $r(i), i = 1, ..., B$, and the sequence of training/desired signals $b(i), i = 1, ... B$ in accordance with the block diagram of Fig. 3. Specifically, at block 28, the Hermitian transpose of the received sample vector is formed and at 30 is multiplied by the received sample vector and a weighting factor of 1 - w. The resulting product is added at 32 to the product formed at 34 by multiplying the weighting factor w by the previous covariance matrix R_{i-1} so as to form the sample covariance matrix R_i for the sequence of received sample vectors $r(1), ..., r(i)$. At 36, the conjugate of the given or desired signal $b(i)$ is formed and at 38 is multiplied by the received sample vector $r(i)$ and the weighting factor $1 - w$. The resulting product is added at 40 to the product of the previous steering vector p_{i-1} and the weighting factor w to form the new steering vector p_i. The steering vector p_i generated from the sequence of given training symbols $b(i), i = 1, ..., B$ or from the sequence of desired filter outputs $b(i), i = 1, ..., B$ as shown in Fig. 3 is an example of an "estimated" steering vector. It should be appreciated that Fig. 3 is only one example of a technique for generating the estimated steering vector. Other known techniques for generating an estimated steering vector can be used as well. A "given" steering vector can be used in accordance with the present invention as well. A "given" steering vector is a steering vector that is already known to the receiver and may be stored, for example, in a memory or buffer of the receiver.

At block 44, the generator 24 generates D + 1 basis vectors v_0 through v_D. The basis vectors are generated in accordance with the diagram depicted in Fig. 4. The first or initial basis vector v_0 is set equal to the steering vector p. Each successive basis vector is formed by multiplying the immediately preceding basis vector by the covariance matrix R so that the basis vectors are generated as follows.

$$v_o = p, \quad v_1 = Rp, \quad v_2 = R^2 p, \ldots, v_{D-1} = R^{D-1} p, \quad v_D = R^D p$$

The generator 24 at block 46 forms the matrix M of basis vectors from the basis vectors v_0 through v_{D-1} so that the matrix M has a N x D dimension. Thereafter, the generator 24 determines the correlation scalars γ_1 through γ_{2D-1} from the basis vectors v_0 through v_D determined at block 44. The correlation scalar γ_{i+j} is generated by multiplying the Hermitian transpose of the basis vector v_i by the basis vector v_j as follows.

$$\gamma_1 = v_0^\dagger v_1, \quad \gamma_2 = v_0^\dagger v_2 \ldots \gamma_D = v_0^\dagger v_D, \quad \gamma_{D+1} = v_D^\dagger v_1, \quad \gamma_{D+2} = v_D^\dagger v_2 \ldots \gamma_{2D-1} = v_D^\dagger v_{D-1}$$

Also $\gamma_0 = v_o^\dagger v_0$ and v_0 is assumed to be normalized to that $\gamma_0 = 1$.

The generator 24 at block 54 forms a correlation matrix Γ from the correlation scalars γ_1 through γ_{2D-1} as follows.

$$\Gamma = \begin{bmatrix} \gamma_1 & \gamma_2 \cdots \gamma_D \\ \gamma_2 & \gamma_3 \cdots \gamma_{D+1} \\ \vdots & \vdots \\ \gamma_D & \gamma_{D+1} \cdots \gamma_{2D-1} \end{bmatrix}$$

At block 54 the generator 24 also generates a reduced rank correlation vector y which is a D x 1 vector formed of the correlation scalars γ_0 through γ_{D-1} as follows.

$$y = \begin{bmatrix} \gamma_0 \\ \gamma_1 \\ \vdots \\ \gamma_{D-1} \end{bmatrix}$$

The generator 24 at block 56 then generates the filter coefficient vector $\tilde{c}$ which is a D x 1 vector generated by solving the following set of linear equations.

$$\Gamma \tilde{c} = y$$

The matrix M of basis vectors formed at block 46 and the filter coefficients $\tilde{c}$ generated at block 56 are applied to the adaptive digital filter 22 so as to generate the reduced rank vector $\tilde{r}(i)$ and the approximate desired signal $\hat{b}(i)$ output from the filter 22 in accordance with blocks 58 and 60. In particular, at block 58, the adaptive digital filter 22 generates the reduced rank vector $\tilde{r}(i)$ by multiplying the Hermitian transpose of the matrix M of basis vectors by the received sample vector of digital data $r(i)$ as follows.

$$\tilde{r}(i) = M^{\dagger} r(i)$$

At block 60, the filter 22 generates the approximate desired signal $\hat{b}(i)$ by multiplying the Hermitian transpose of the D x 1 filter coefficient vector $\tilde{c}$ by the reduced rank vector $\tilde{r}(i)$ as follows.

$$\hat{b}(i) = \tilde{c}^{\dagger} \tilde{r}(i)$$

It has been found that the reduced rank adaptive digital filter described above achieves near full rank performance with a value of D equal to 8 or less for a Direct Sequence (DS) Code Division Multiple Access (CDMA) communication system. For other applications, the value of D can be determined in accordance with the method depicted in Fig. 5. As shown in Fig. 5, at block 60 an index n is initialized to 1. Thereafter, at block 62 the nth basis vector v_n is generated by

multiplying the previous basis vector v_{n-1} by the covariance matrix R where the first basis vector is the steering vector p so that

$$v_n = R^n p$$

as discussed above. Thereafter, a matrix M_{n-1} of basis vectors is formed at block 64 as follows.

$$M_{n-1} = [v_0, v_1, \dots v_{n-1}]$$

At block 66, the orthogonal projection u_n, which is a N x 1 vector, is generated as follows.

$$u_n = v_n - M_{n-1}(M^\dagger_{n-1}M_{n-1})^{-1}M_{n-1}^\dagger v_n$$

The respective lengths l and g of the vectors u_n and v_n are computed at block 67 in accordance with the following equations.

$$l = \|u_n\|$$

$$g = \|v_n\|$$

where l and g are scalars. Thereafter, at block 68, it is determined whether the length l divided by the length g is greater than a small constant δ, where δ may be set equal to 0.01 for example. If so, n is incremented by 1 and steps 62 through 68 are repeated. Steps 62 through 70 are repeated until l divided by g is not greater than δ for a given value of n. At that point, D is set equal to n - 1.

In an alternative method, the value of D is chosen to minimize the "*a posteriori* Least Squares cost function C" where

$$C = \sum_{m=m_0}^{i}|b(m) - \tilde{c}^\dagger_{D}\tilde{r}(m)|^2$$

where $\tilde{c}_D$ is the reduced rank filter coefficient vector generated from the sequence of $b(1), \dots b(B)$ and the sequence of $r(1), \dots r(B)$ and $B < m_o$.

The reduced rank adaptive filter of the present invention projects the N x 1 received signal vector $r(i)$ onto the lower dimensional

subspace which results in a D x 1 vector $\tilde{r}(i)$ where D can be much less than N while still allowing the filter 10 to achieve near full rank performance. For interference suppression applications, the optimum filter rank D does not substantially increase with the dimension of the signal subspace N. This enables much faster tracking and convergence than can be achieved with prior techniques. Furthermore, the method of the present invention does not rely on an estimate of the signal space via, for example, eigen-decomposition so that the complexity of the present method is substantially reduced. Although the method was described in detail for a wireless communication system application, it can be used in any adaptive filtering application including, echo and noise cancellation, channel equalization, radar processing, adaptive antennae arrays, interference suppression, etc.

Many modifications and variations of the present invention are possible in light of the above teachings. Thus, it is to be understood that, within the scope of the appended claims, the invention may be practiced otherwise than as described hereinabove.

What is claimed and desired to be secured by Letters Patent is:

- 11 -

CLAIMS

1.   A method of filtering successive received signal samples, a group of N successive samples forming a received sample vector of digital data having a N x 1 dimension to provide an approximate desired signal comprising:

generating a set of basis vectors where each successive basis vector is a function of a given or an estimated steering vector and successively greater powers of a covariance matrix for a sequence of received sample vectors of data with the initial basis vector being formed from the steering vector;

generating a reduced rank vector of digital data having a D x 1 dimension, where D is less than N, from a matrix of D basis vectors and a received sample vector of data;

generating a D x 1 filter coefficient vector from the generated basis vectors; and

generating the approximate desired signal from the filter coefficients and the reduced rank vector of data.

2.   A method of filtering successive received signal samples as recited in claim 1 where D + 1 successive basis vectors are generated by multiplying an immediately preceding basis vector of data by the covariance matrix for the received sample vector.

3.   A method of filtering successive received signal samples as recited in claim 1 wherein the reduced rank vector is generated by multiplying the Hermitian transpose of the matrix of basis vectors by the received sample vector of data.

4.   A method of filtering successive received signal samples as recited in claim 1 including forming a D x 1 correlation vector with

correlation scalars, each correlation scaler γ_i formed by multiplying the Hermitian transpose of the initial basis vector by the ith basis vector for

5   i equal 0 through D-1.

5.   A method of filtering successive received signal samples as recited in claim 4 including forming a D x D correlation matrix Γ from 2D-1 correlation scalars generated from pairs of the basis vectors and generating a D x 1 vector of filter coefficients $\tilde{c}$ by solving a set of

5   linear equations defined by $\Gamma\tilde{c} = y$ where y is the correlation vector.

6.   A method of filtering successive received signal samples as recited in claim 1 including generating the desired signal by multiplying a Hermitian transpose of the filter coefficient matrix by the reduced rank vector of data.

7.   A method of filtering successive received signal samples, a group of N successive samples forming a received sample vector of digital data having a N x 1 dimension to provide an approximate desired signal comprising:

5   determining a received sample covariance matrix from a sequence of received sample vectors of digital data;

determining a steering vector of data;

generating a reduced rank vector of digital data having a D x 1 dimension where D is less than N from the received sample vector of

10   data and a matrix of D basis vectors of data wherein each successive basis vector of data is generated by multiplying an immediately preceding basis vector of data by the covariance matrix and the initial basis vector is formed from the steering vector of data;

generating D filter coefficients from a plurality of correlations

15   between pairs of basis vectors of data; and

generating the approximate desired signal from the filter coefficients and the reduced rank vector of data.

8.   A method of filtering successive received signal samples as recited in claim 7 wherein the reduced rank vector is generated by multiplying the Hermitian transpose of the matrix of basis vectors by the received sample vector of data.

9.   A method of filtering successive received signal samples as recited in claim 7 including forming a D x 1 correlation vector with correlation scalars, each correlation scaler γ_i formed by multiplying the Hermitian transpose of the initial basis vector by the ith basis vector for
5   i equal 0 through D-1.

10.   A method of filtering successive received signal samples as recited in claim 9 including forming a D x D correlation matrix Γ from 2D-1 correlation scalars generated from pairs of the basis vectors and generating a D x 1 vector of filter coefficients $\tilde{c}$ by solving a set of
5   linear equations defined by $\Gamma\tilde{c} = y$ where y is the correlation vector.

11.   A method of filtering successive received signal samples as recited in claim 7 including generating the desired signal by multiplying a Hermitian transpose of the filter coefficient matrix by the reduced rank vector of data.

12.   A method of filtering successive received signal samples, a group of N successive samples forming a received sample vector of digital data having a N x 1 dimension to provide an approximate desired signal comprising:

5   generating $D + 1$ basis vectors wherein D is less than N and each successive basis vector is a function of a given or an estimated steering vector and successively greater powers of a covariance matrix for a sequence of the received sample vectors of data with the first basis vector being formed from the steering vector;

10   generating a reduced rank vector of digital data having a $D \times 1$ dimension from a matrix of D basis vectors and a received sample vector of data;

generating $2D - 1$ correlations between pairs of basis vectors;

generating D filter coefficients from a matrix of said correlations

15 and a vector of D correlations; and

generating the approximate desired signal from the filter coefficients and the reduced rank vector of data.

13.   A method of filtering successive received signal samples as recited in claim 12 where $D + 1$ successive basis vectors are generated by multiplying an immediately preceding basis vector of data by the covariance matrix for the received sample vector.

14.   A method of filtering successive received signal samples as recited in claim 12 wherein the reduced rank vector is generated by multiplying the Hermitian transpose of the matrix of basis vectors by the received sample vector of data.

15.   A method of filtering successive received signal samples as recited in claim 12 including forming a $D \times 1$ correlation vector with correlation scalars, each correlation scaler γ_i formed by multiplying the Hermitian transpose of the initial basis vector by the ith basis vector for

5   i equal 0 through D-1.

16. A method of filtering successive received signal samples as recited in claim 15 including forming a D x D correlation matrix Γ from 2D-1 correlation scalars generated from pairs of the basis vectors and generating a D x 1 vector of filter coefficients $\tilde{c}$ by solving a set of linear equations defined by $\Gamma\tilde{c} = y$ where y is the correlation vector.

17. A method of filtering successive received signal samples as recited in claim 12 including generating the desired signal by multiplying a Hermitian transpose of the filter coefficient matrix by the reduced rank vector of data.

18. A method of filtering successive received signal samples, a group of N successive samples forming a received sample vector of digital data having a N x 1 dimension to provide an approximate desired signal comprising:

selecting a dimension D for a reduced rank vector onto which a received sample vector of data is projected, where D is less than N;

generating a set of basis vectors wherein each successive basis vector is a function of a given or an estimated steering vector and successively greater powers of a covariance matrix a sequence of received sample vectors of data with the first basis vector being formed from the steering vector;

generating a reduced rank vector of digital data having a D x 1 dimension from a matrix of D basis vectors and the received sample vector of data;

generating D filter coefficients from the generated basis vectors; and

generating the approximate desired signal from the filter coefficients and the reduced rank vector of data.

- 16 -

19. A method of filtering successive received signal samples as recited in claim 18 where D is selected to be less than or equal to 8.

20. A method of filtering successive received signal samples as recited in claim 18 where D is selected to minimize an *a posteriori* Least Squares cost function.

21. A method of filtering successive received signal samples as recited in claim 20 where $D + 1$ successive basis vectors are generated by multiplying an immediately preceding basis vector of data by the covariance matrix for the received sample vector.

22. A method of filtering successive received signal samples as recited in claim 20 wherein the reduced rank vector is generated by multiplying the Hermitian transpose of the matrix of basis vectors by the received sample vector of data.

23. A method of filtering successive received signal samples as recited in claim 20 including forming a D x 1 correlation vector with correlation scalars, each correlation scaler γ_i formed by multiplying the Hermitian transpose of the initial basis vector by the ith basis vector for
5   i equal 0 through D-1.

24. A method of filtering successive received signal samples as recited in claim 23 including forming a D x D correlation matrix Γ from 2D-1 correlation scalars generated from pairs of the basis vectors and generating a D x 1 vector of filter coefficients $\tilde{c}$ by solving a set of
5   linear equations defined by $\Gamma\tilde{c} = y$ where y is the correlation vector.

- 17 -

25.   A method of filtering successive received signal samples as recited in claim 20 including generating the desired signal by multiplying a Hermitian transpose of the filter coefficient matrix by the reduced rank vector of data.

ABSTRACT OF THE DISCLOSURE

A reduced rank adaptive digital filtering method is described for a received signal consisting of a sequence of N x 1 received vectors. Each received vector is formed from a group of N successive samples. D + 1 basis vectors are generated where D is less than N and the dimension of a desired reduced rank subspace. Each successive basis vector is generated by multiplying an immediate preceding basis vector by the covariance matrix for the sequence of received sample vectors and the first basis vector is formed from a given or estimated steering vector. D filter coefficients are generated from correlations between pairs of basis vectors. The adaptive digital filter of the present invention achieves near optimal rank performance with substantially fewer training symbols than heretofore possible.

FIG 1

$r(i)$ $b(i)$

COMPUTE COVARIANCE MATRIX R AND STEERING VECTOR P FROM OBSERVATION SIGNAL AND TRAINING/DESIRED SIGNAL ~26

COMPUTE BASIS VECTORS $V_0, V_1, V_2, \ldots, V_D$ ~44

FORM MATRIX OF BASIS VECTORS $M = [P \ V_1 \ V_2 \cdots V_{D-1}]$ ~46

COMPUTE CORRELATION SCALERS $\gamma_1, \gamma_2, \ldots, \gamma_{2D-1}$ ~52

FORM CORRELATION MATRIX Γ AND REDUCED RANK CORRELATION VECTOR y ~54

COMPUTE FILTER COEFFICIENTS $\tilde{c}$; $\tilde{c} = \Gamma^{-1} y$ ~56

OUTPUT C

$r(i)$

COMPUTE REDUCED RANK VECTOR $\tilde{r}(i) = M^{\dagger} r(i)$ ~58

COMPUTE FILTER OUTPUT $b(i) = \tilde{c}^{\dagger} \tilde{r}(i)$ ~60

OUTPUT ESTIMATES OF $b(i)$

FIG 2

$(1-\omega)$

$r(i)$ ⊗ 30 ⊕ 32 → R_i

28

HERMITIAN
TRANSPOSE

$r^\dagger(i)$

⊗ 34 ω

R_{i-1}

$b(i)$ → CONJUGATE → ⊗ 38 → ⊕ 40 → P_i

36 $(1-\omega)$

⊗ 42 ω

P_{i-1}

FIG 3

$V_0 = P$ ⊗ 47 V_1 ⊗ 48 V_2 ⊗ 49 V_3 ... ⊗ 50 V_D →

R R R R

FIG 4

FIG 5

SET $n = 1$ ~ 60

GENERATE BASIS ~ 62
VECTOR: $v_n = R v_{n-1}$

FORM MATRIX M_{n-1}: ~ 64
$M_{n-1} = [v_0, v_1, \cdots, v_{n-1}]$

DETERMINE ORTHOGONAL PROJECTION
$u_n = v_n - M_{n-1}(M_{n-1}^\dagger M_{n-1})^{-1} M_{n-1}^\dagger v_n$ ~ 66

DETERMINE LENGTHS
$l = \|u_n\|$, $g = \|v_n\|$ ~ 67

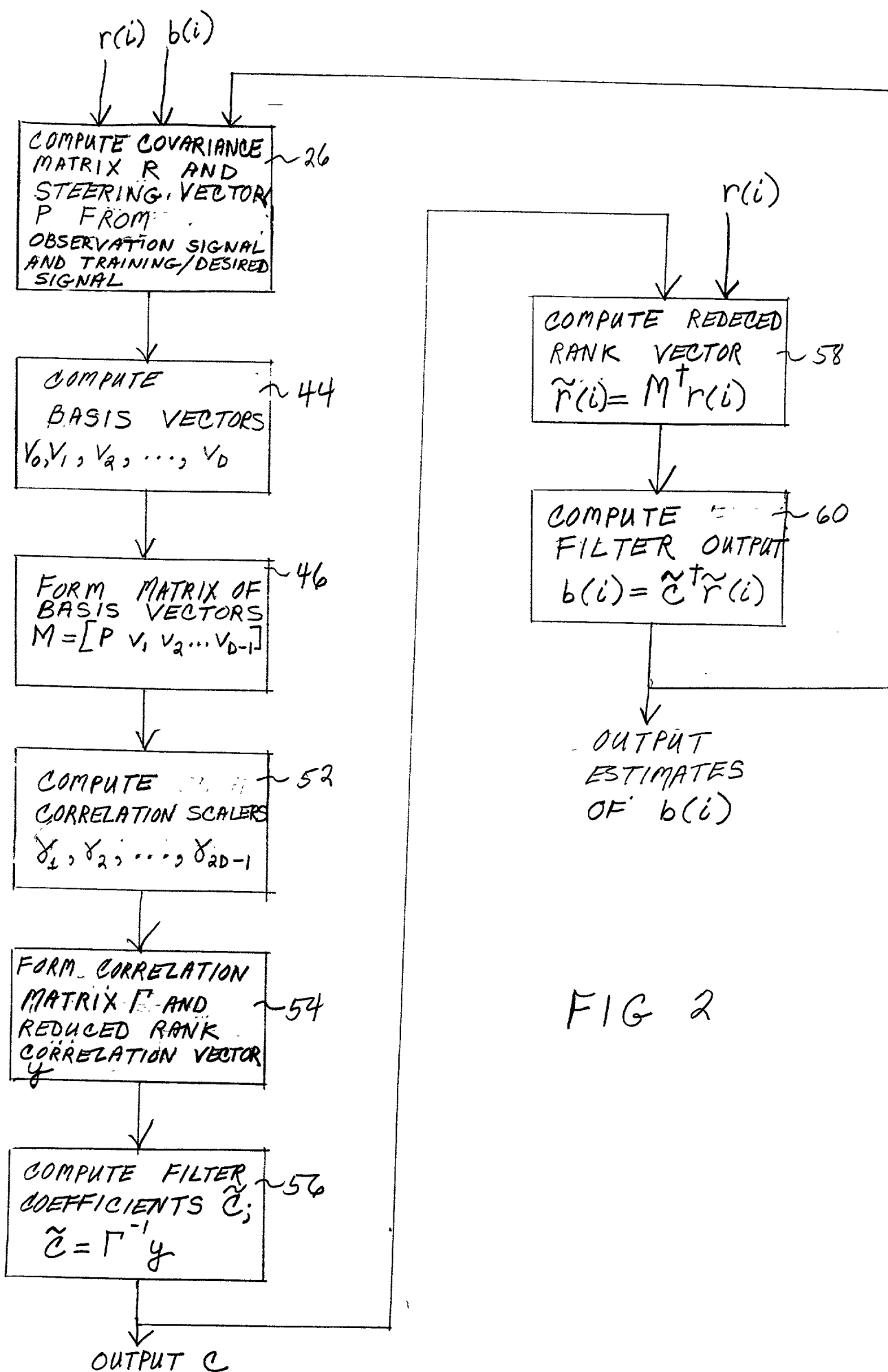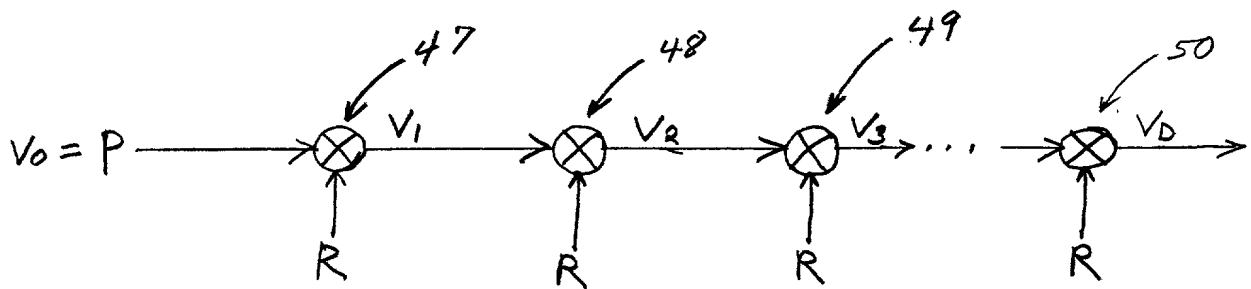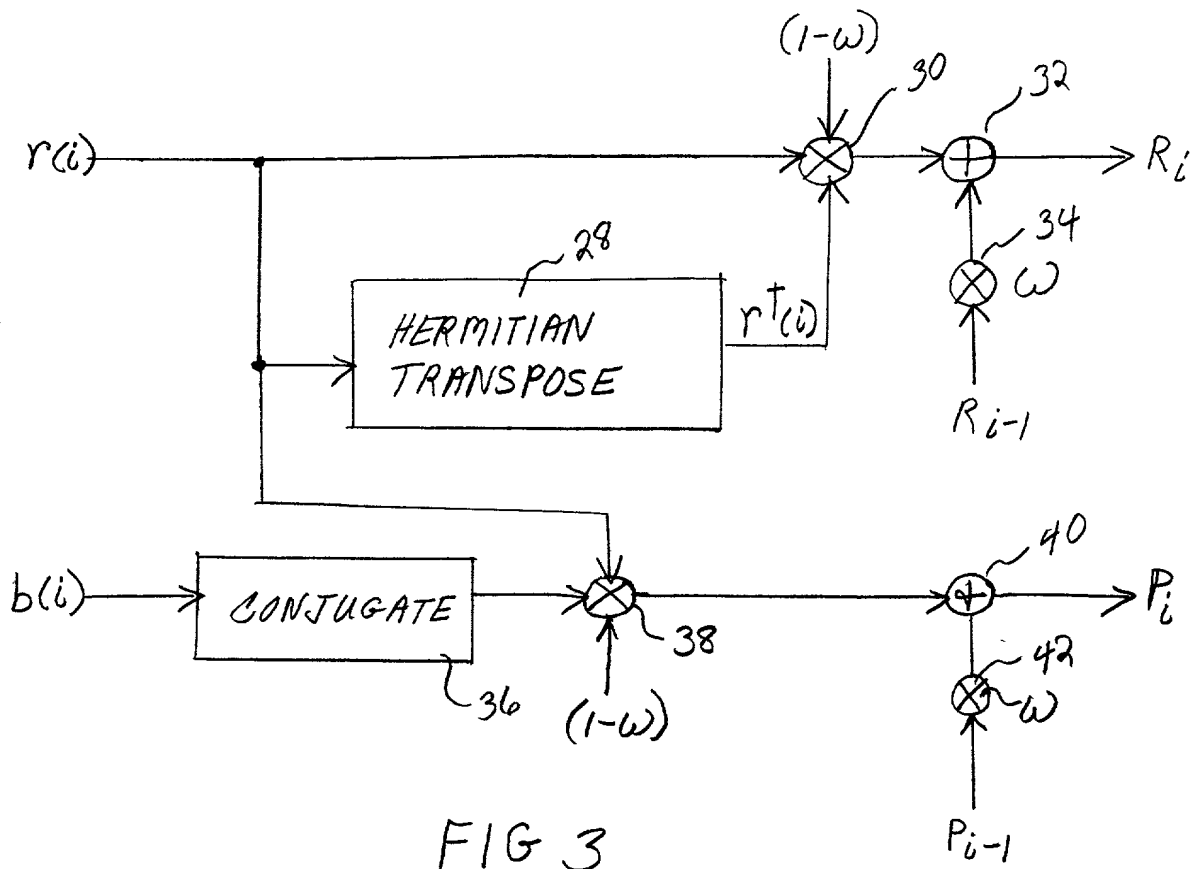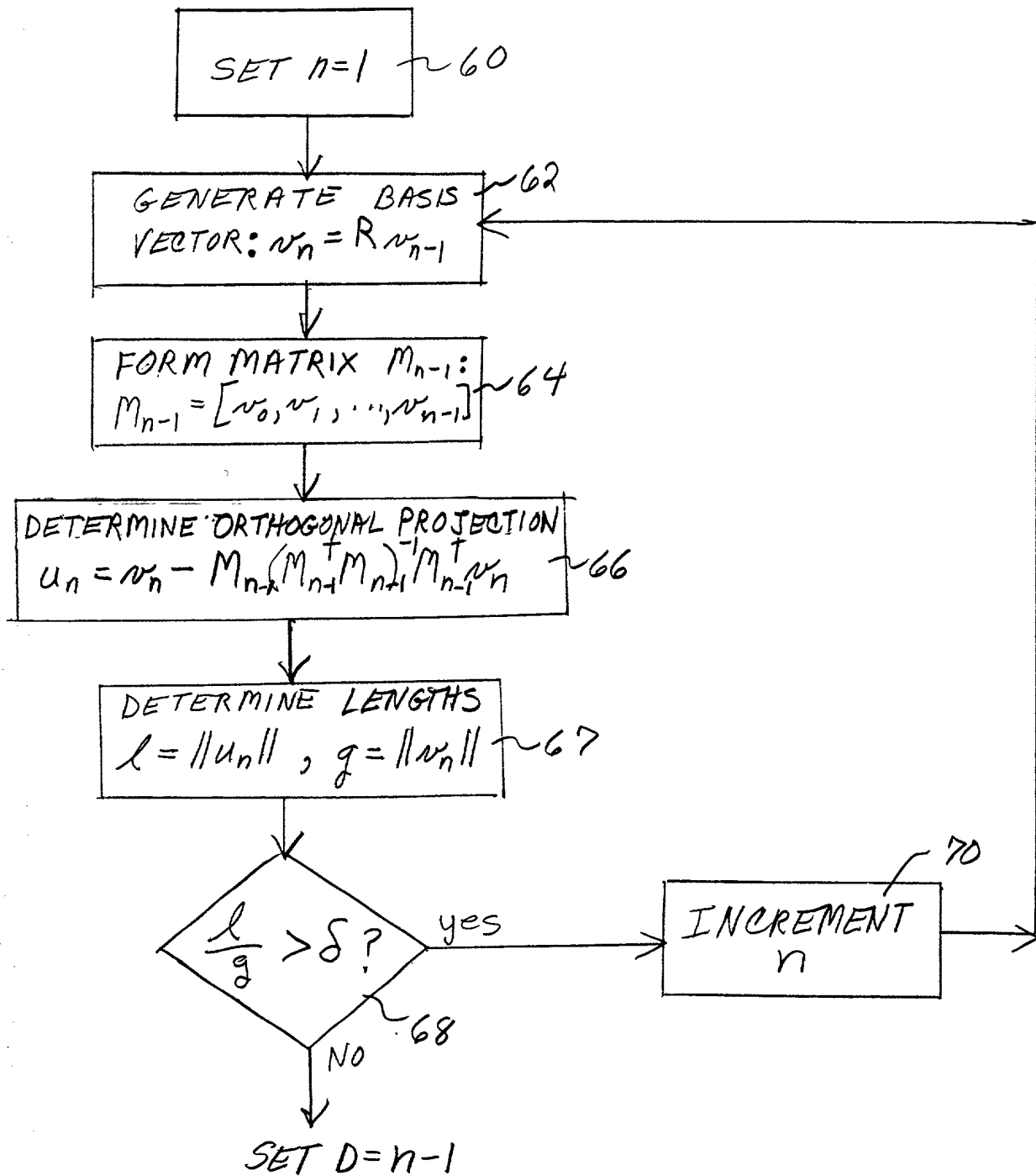$\dfrac{l}{g} > \delta$?   — yes →   INCREMENT n ~ 70

~ 68

NO

SET $D = n - 1$

DECLARATION AND POWER OF ATTORNEY
(Case No. 12569US01)

As the below-named inventors, we hereby declare that:
our residence, post office address, and citizenship are as stated below next to our names.

We believe we are the original, first and joint inventors of the subject matter which is claimed and for which a patent is sought on the invention entitled

REDUCED RANK ADAPTIVE FILTER

the specification of which is attached hereto.

We hereby state that we have reviewed and understand the contents of the above-identified specification, including the claims.

We acknowledge the duty to disclose information which is material to the examination of this application in accordance with Title 37, Code of Federal Regulations, Section 1.56(a).

And we hereby appoint:

PTO REGISTRATION NUMBERS

George P. McAndrews	Reg. No. 22,760	Donald J. Pochopien	Reg. No. 32,167
John J. Held	Reg. No. 21,061	Sharon A. Hwang	Reg. No. 39,717
Timothy J. Malloy	Reg. No. 25,600	David D. Headrick	Reg. No. 40,642
William M. Wesley	Reg. No. 26,521	Dean D. Small	Reg. No. 34,730
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Gregory C. Schodde	Reg. No. 36,668		
Edward W. Remus	Reg. No. 25,703		

the address and telephone number of each of whom is McAndrews, Held & Malloy, Ltd., 500 West Madison Street, 34th Floor, Chicago, Illinois 60661, 312-775-8000, as our attorneys with full power of substitution and revocation to prosecute this application and to transact all business in the Patent and Trademark Office connected therewith.

We hereby declare that all statements made herein of our own knowledge are true and that all statements made on information and belief are believed to be true; and further that these statements were made with the knowledge that willful false statements and the like so made are punishable by fine or imprisonment, or both, under Section 1001 of Title 18 of the United States Code and that such willful false statements may jeopardize the validity of the application or any patent issued thereon.

Full name of inventor_____Michael L. Honig_____

Inventor's signature_____ Date_____

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2